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In acoustic thermometry of the ocean, interpretation of the results of the measurements implies that all distances between sources and receivers in the thermometry system are fixed during an experiment. This means that the acoustic sources and receivers should be firmly installed on the bottom. However, it is not always possible to deploy such stable systems in the ocean, especially in deep water. One of the acoustic emitting complexes for the ACOUS (Arctic Climate Observation using Underwater Sound) experiment is planned for deployment in the deep-water Central Arctic Basin. Arctic conditions make it difficult to deploy a long-baseline system, since the ice drift cannot be controlled and the navigation beacons cannot be accurately deployed at the determined points on the bottom around the main mooring system. In these conditions, the beacons should be self-locating and the system construction adapted for fast deployment. Moreover, the necessary resolution of the thermometry system in the ACOUS experiment requires positioning accuracy better than that in the available commercial devices. This paper discusses these requirements of the acoustic positioning system as well as addressing the issue of communication between the complex and the shore.

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DEVELOPMENT OF THE METHOD FOR DEPLOYMENT OF AN AUTONOMOUS SOURCE FOR YEAR-ROUND ACOUSTIC MONITORING OF THE ARCTIC OCEAN THE ACOUSTIC COMPLEX FOR HORIZONTAL LOCATION OF AN AUTONOMOUS SOURCE AND COMMUNICATION OF THE NAVIGATION DATA

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INTRODUCTION

In acoustic thermometry of the ocean, interpretation of the results of the measurements implies that all distances between sources and receivers in the thermometry system are fixed during the experiment. This means that the acoustic sources and receivers should be firmly installed on the bottom. However, it is not always possible to deploy such stable systems in the ocean, especially in deep water. Usually, underwater acoustic systems are deployed in the ocean by means of flexible mooring with a rope, or a cable. In some cases, an acceptable spatial stability of the mooring system can be achieved by the use of a large float, which would put sufficient tension on the rope. If the mooring length is too long and the water current too fast and changeable, an ordinary single-mooring system would not provide the stability required, since the tension of the mooring cable is restricted by its destruction limit. In that case, spatial deviation of the mooring system should be tracked by a special navigation system.

One of the acoustic emitting complexes for the ACOUS (Arctic Climate Observation using Underwater Sound) experiment is planned for deployment in the deep-water Central Arctic Basin. As shown in [1], the mooring system carrying that complex, should be provided with a long-baseline acoustic positioning system for locating horizontal deviations of the source. Many commercial systems are available for acoustic navigation of underwater devices, but very few of these systems are adapted for installation in the Arctic Ocean, from the drifting ice. Arctic conditions make it difficult to deploy a long-baseline system, since the ice drift cannot be controlled and the navigation beacons cannot be accurately deployed at the determined points on the bottom, around the main mooring system. In these conditions, the beacons should be self-locating and the system construction adapted for fast deployment. Moreover, the necessary resolution of the thermometry system in the ACOUS experiment requires positioning accuracy better than that in the available commercial devices. All of these requirements have been taken into account in designing the acoustic positioning system considered in the first section of the report.

Another problem concerns the communication of the positioning data from the emitting complex to the shore. Since the complex should be autonomous, the data cannot be transmitted through a cable. Therefore, we have only two ways to save the navigation information - either to accumulate the data in a special storage device in the complex, or to transmit these data with the low-frequency signals intended for the tomographic measurements. Use of the first (the traditional) way, in the proposed system, has two fundamental disadvantages. Firstly, we would not have access to the positioning data in real time, and therefore could not interpret the acoustic thermometry data during the long experiment, and, secondly, we would, without fail, have to recover the emitting complex

after the experiment. However, this is a very difficult, as well as expensive task, especially in the Central Arctic covered year-round with ice.

The second way seems to be much more appropriate for our application, since, for communication, it implies using the same acoustic channel as that intended for the thermometry measurements. The principles of organization of a communication channel incorporated into an acoustic thermometry system, and the method of signal processing common to both tomographic measurements and decoding of communication information, are considered in the second section of the report.

I. THE ACOUSTIC POSITIONING SYSTEM FOR AN AUTONOMOUS ACOUSTIC SOURCE IN THE ACOUS EXPERIMENT

It has been proposed that the acoustic source for the ACOUS experiment be installed at an autonomous buoy station (ABS). This station will be subject to horizontal deviations due to changes in the water current. These uncontrolled deviations will produce errors in the measurements of the signal propagation time in the acoustic tomography system. Therefore, it is necessary to track horizontal deviations of the source during the experiment. Variations in the water current due, mostly, to tidal motion, are relatively slow, therefore the horizontal position of the source can be tracked once per 1 hour acoustic transmission.

The Acoustic Positioning System (APS) described below, is intended to monitor the relative horizontal location of the source and to prepare the navigation data for long-distance transmission to the receiving station with a low-frequency tomographic signal.

Initial data

1. **<u>Source location</u>**

- Deployment of the acoustic source (S) is planned for the northern part of the St. Anna Strait. The depth in this region is 600±50m. The ice cover is expected to be continuous in winter, and have a thickness of up to 4 m. The speed of the ice drift is, generally, no more than 0.1 m/sec, but sometimes can reach 0.3 m/sec. Experimental data on the water current in the proposed region are very approximate. We assume that the current speed may be as much as 0.2 m/sec in some water layers.
- The sound speed profile and its seasonal variability has been taken from the hydrologic data collected from numerous experimental measurements in the Northern

Kara Sea. The widely differing values of the sound speed observed at various depths in this region are shown in Fig. 1.

- The bottom relief is expected to be smooth and gently sloping. The following upper limits have been assumed in designing the APS: typical slope - about 30 m in depth per 1000 m in range, local roughness - not higher than 5 m.

2. Mooring

The assumed locations of the ABS components are given in the following table:

depth, m	ABS component
50 ±20	buoy
100 ±20	Source, other equipment
600 ±20	anchor

3. **Requirements in designing an APS**:

- Accuracy of measurements of the horizontal deviations $|\Delta r|$ of the source - ±2.5 m;
- Presentation of the positioning data: serial binary code of 18 bit maximal length;
- Battery life: not less 12 months and 300 complete position soundings ;
- Deployment: from an ice floe, or from an icebreaker and surrounding ice floes;
- Life time: not less 12 months;
- Possibility of updating the system for a depth up to 2000 m.

General description

The Acoustic Positioning System (APS) is intended to measure coordinates X, Y of the source, relative to the array of bottom-mounted acoustic beacons-transponders (long-baseline system).

The source position is determined from the measured slanting ranges, i.e. from the travel times of acoustic pulses transmitted from the transceiver located at the top of the

ABS (near the acoustic source), to the transponders, and back from the transponders to the transceiver.

The proposed system has as a distinguishing feature, compared to similar commercial devices, the capability of completely autonomous work. This implies autonomous calibration of the relative spatial location, as well as autonomous measurements of the horizontal deviations.

The APS contains the components shown in Fig. 2. Three bottom-mounted intelligent beacon-transponders (T) are located around the ABS at the vertices of an approximately equilateral triangle with distances to the ABS anchor of 500-600 m. In addition, the same beacon is deployed at the ABS anchor. The transceiver MIR (Module of Interrogator-Receiver) is attached to the line of the ABS near the source. The MIR provides activation of the beacons, and measurements of the slant ranges. Final data processing and general system management are assumed to be a function of the Main Control Unit (MCU) which controls operation of the acoustic source, and generates low-frequency phase-coded signals for long-distance transmissions.

The procedure of positioning involves: (1) determination of the baseline distances between the beacons, (2) measurements of the beacon depths via pressure sensors, (3) measurements of the slant ranges, and (4) calculation of the horizontal coordinates of the source. The procedure starts with a command from the MCU which directs the MIR to execute sequential acoustic interrogation of the beacons in either the calibrating or measuring modes. The interrogation code includes a common part which controls switching of all beacons from a STAND-BY mode to an active regime, followed by a 5-bit part containing the beacon number and the index of the program to be executed. The regular measuring mode provides only for determination of the MIR-T_i-MIR travel times.

In the calibrating mode, the APS executes the following additional procedure: the beacon designated as Ti recognizes its calibrating command in the program number of the MIR signal, and then interrogates other beacons Tj. Beacons Tj respond to this interrogation sending the return signals to the MIR. The MIR receives the replies from Tj, which permits the measurements of travel times MIR-Ti-Tj-MIR. After completion of the calibration cycle, the MIR will be provided with enough data for determination of the baselines and the slant ranges. For telemetry of the actual beacon depths in the calibrating cycle, the MIR interrogates, each beacon, sequentially, with a special command. Each beacon Ti replies with a signal which carries the encoded beacon depth measured by a built-in pressure sensor. The results of the calibration are fed into the MCU for further processing.

The APS data are proposed to be transmitted to the acoustic receiving arrays, and further - to the shore, via coding of low-frequency tomographic signals (see Section II in this Report). Two different ways of representation of the APS data could be applied for transfer to the receiving system. This could be either two horizontal coordinates (X, Y), or three travel time meanings (T1, T2, T3). In the second case one can use only 6 lower bits of the APS data for the transmission of T1, T2, and T3 with a 18-bit code.

Module of Interrogator and Receiver (MIR)

The MIR sends ultrasonic interrogations to the beacons, receives their replies, measures the slant ranges and transmits the data to the MCU.

The functional scheme of MIR is shown in Fig. 3. The MIR electronics consist of the following units: a microprocessor-Controller (C), a Time-to-Digit Converter (TDC), a hydroacoustic Receiver ® and an output power Amplifier (A). The receiver contains: a hydrophone, a band-pass amplifier, a comparator, and a pulse-duration discriminator.

The controller executes the program of APS operations stored in the ROM, pre-processing of the positioning data, and command/data exchange with the MCU computer. Its design employs a microprocessor of the type P1885AH-2 (RAM - 8 kB, ROM - 1 kB, number of unit addresses - 32).

It is essential that the parameters of the input circuits of the receiver be optimized to provide sufficient accuracy in travel time measurements. To prevent extension of the pulse front, the pass band of the receiver is chosen to be reasonably wide, being approximately 2000 Hz. It allows us to reduce the sensitivity of travel time measurements to the influence of stochastic deviations of the comparator threshold, and variations in the signal amplitude. To hold the signal-to-noise ratio (SNR) at an optimal level, we introduce a pulse-duration selector into the discriminator. In addition to this, the controller carries out regulation of a linear gate scheme by switching the receiver input on, at the moment of 200 msec before the arrival of the beacon reply.

The transmission carrier frequency of the MIR (and transponders) has been chosen to serve as a compromise between the different factors which affect the APS capability. On the one hand, the APS resolution improves with an increase in the carrier frequency while the noise protection, also, improves with a shortening of the measurement cycle. On the other hand, the signal attenuation increases with an increase in frequency, which leads to a decline of the SNR. There are a number of different recommendations on the choice of the optimal frequency for long-baseline systems. For estimation of the optimal frequency for measurements of slant range r, one could use, for example, the following formula [1]:

$$F[\text{kHz}] = \frac{39}{r[\text{km}]^{\frac{2}{3}}}$$

Thus, for slant distances of 0.9 km and 2.5 km (for sea depths of 0.6 km and 1.5 km) one could choose frequencies of 42 kHz and 20 kHz respectively. Taking into account an increased attenuation of high-frequency acoustic signals in the cold water of the Arctic

Ocean (see Fig.7), we propose to retain the same carrier frequencies, as are used in the Baikal positioning system: 32.768 kHz - for the MIR interrogation, and 28 kHz - for the beacon reply. A similar positioning system deployed at the bottom of Lake Baikal (at a depth of 2 km) for tracking the Baikal Neutrino telescope, has provided reliable navigation measurements for several years, though it operates in fresh water [2].

Intelligent Beacon-Transponder

The bottom transponders operate as bench-marks in the acoustic coordinate grid. We propose to use direct baseline measurements to determine the relative location of the beacons.

We propose to deploy the beacons on to the bottom, from those points on the ice floe which make the distances from the beacons to the ABS anchor approximately equal to the height of MIR above the bottom. With such a system geometry, the accuracy of horizontal positioning would be optimally matched with stable reception of the acoustic navigation signals.

We plan to install the beacons on a single mooring system (Fig.10). The beacon hydrophones cannot be deployed at too great a height above the bottom, since horizontal deviations of their positions due to changes in the water current could degrade the accuracy of the acoustic navigation measurements. On the other hand, one cannot deploy the beacons too close to the bottom, because, an acoustic ray propagated near the bottom could undergo both interaction with the bottom, due to ray refraction in an inhomogenous water column, and reflection from bottom roughnesses. In our design of the beacons, we propose to deploy the hydrophones 8 - 10 meters above the bottom.

In any case, the direct ray should interfere with the bottom reflected ray. To achieve reliable triggering of the transponder and diminish the effects of ray interference,

we propose to install two hydrophones on each beacon with a vertical offset of about 1 m. For the APS geometry proposed, the baseline is about 850 m, and the difference in length of the bottom reflected rays, aimed at two vertically separated hydrophones, is about half a wavelength, while the difference in length of the direct rays is negligible relative to the wavelength. In this case, when one of the hydrophones is situated at the interference minimum, the other one should be at the maximum. In such a system, the amplitude of the interference signal on one of the hydrophones in the pair would be not less than the amplitude of the direct signal. The selection of the hydrophone in each pair, on each beacon, should be made during the calibrating regime.

The functional scheme of the beacon-transponder is shown in Fig. 4. It is similar to that of the MIR and contains a two-channel Receiver (R), a microprocessor Controller (C), an output power Amplifier (A), components of a Pressure telemetry Channel (PC) and a Power Supply (PS). Each receiver channel consists of a hydrophone, a band-pass amplifier, a comparator, and a pulse-duration discriminator. To protect the receiver from noise, the beacon controller performs a pulse-duration selection similar to that in the MIR.

The beacon spends most of the time in a stand-by mode, when the controller, the output amplifier, and the pressure channel are turned off. To activate the transponder, the MIR sends an encoded command the time diagram of which is shown in Fig. 5. The first double-length pulse switches all of the beacons from the stand-by mode to the regular operating mode. Each beacon controller analyses the farthest pulse sequence to identify the number of the beacon interrogated, and the number of the operating program. If the controller does not receive the next interrogation within a specific time period, the built-in clock turns the beacon on to the stand-by mode.

As shown in Fig.5, the pulse code transmitted from the MIR to the beacons, consists of the 2 msec activating pulse and 5 (or more) 1 msec command pulses, separated by 2 msec intervals. The pulses are filled up by the carrier frequency of 32.768

kHz. In the serial digital code, each 1 msec pulse is interpreted as 1, and each 1 msec gap - as 0. The complete digital set consists of 32 combinations.

The beacons perform the following three operation algorithms:

1. Measurement of slant distances.

   The MIR sends the interrogation corresponding to the beacon, and starts the time calculation. The MIR receiver should be switched on 200 msec before the theoretical arrival time of the beacon's reply. When the interrogated beacon has recognized the command, it responds by a single pulse of 1.5 msec in length at a carrier frequency of 28 kHz. This reply stops time counting in the MIR.
2. Calibration of the beacons array.

   The MIR sends the command and starts counting the time. The interrogated beacon replies with the same code as for the slant range measurement (f = 32.768 kHz). This signal initiates the second indicated beacon, while the MIR ignores it. Time counting in the MIR would be stopped only with the arrival of the standard reply of the second beacon (1.5 msec pulse at 28 kHz).
3. Pressure telemetry.

   The MIR sends the special command for pressure measurements and sets the TDC into a state of readiness to start the measurements. The interrogated beacon responds to the MIR by the pulse which starts the time counting. Simultaneously, the beacon controller starts the counting of the pulses from the pulse converter of the pressure sensor. The number of pulses counted controls the time interval between the first and second responses from the beacon. This interval relates to the measured depth as 1 msec per 1 m of the depth. The acoustic pulse of the second response from the beacon interrupts the time counting in the MIR.

Accuracy of the Measurements

Three major factors influence the accuracy of navigation measurements in a long-baseline positioning system: characteristics of the medium, quality of calibration, and instrumental accuracy.

1. Characteristics of the Medium.

Spatial and temporal variations of the sound speed in the water column near the APS can cause the greatest uncertainties in the measured value of the source position. In the ideal case of absolutely symmetrical deployment of the beacons relative to a central point, located at the mooring anchor, horizontal positioning in the APS should be insensitive to temporal variations in the sound speed in a horizontally stratified water column. In the case of asymmetrical deployment of the beacons, with a 100 m offset of the array center relative to the mooring anchor, the error in horizontal positioning of the source due to possible changes in the sound speed profile (see Fig.1) should not exceed 0.5 m.

The navigation errors become more significant, if the sound speed profile varies within the APS horizontal direction. For estimation of possible errors, we assumed that the sound speed profile was subject to changes only along one of three slant distances of a symmetrical system. Figure 6 shows the error of horizontal positioning as a function of deviations of the mean sound speed along the slant trajectory T-MIR. The maximal variation in the mean sound speed of ±3.5 m/sec corresponds to extreme changes in the sound speed profile (Fig.1). Such changes at only one side of the beacon's array are more hypothetical than realistic. Even in this improbable case, the error in horizontal positioning of the source would not exceed 1.5 m.

The propagating acoustic ray undergoes refraction due to the horizontal stratification of the sound speed. For the slant trajectories MIR-T-MIR, the refraction could lead to an increase in the acoustic travel time by a negligible value of 5 μsec, which results in an error of a few cms in positioning. The effects of the refraction of the rays T-T, propagated close to the bottom, can be much more significant. They restrain, mainly, the range of direct acoustic propagation above the bottom. Taking the vertical gradient of the sound speed near the bottom from experimental profiles, we estimated the maximal span of the beacons which allows direct propagation of the signal over a flat bottom. For the beacon's hydrophones deployed at 10 m above the bottom, the range of reliable transmissions between the beacons does not exceed 2 km.

Acoustic propagation losses and the acoustic noise in the ocean determine the requirements for the power level of the high-frequency sources in the APS. Figure 7 demonstrates the difference in the attenuation of acoustic signals at 30 kHz in cold fresh, and in sea water. The difference achieves 20 dB at distances of 2 km.

The ocean noise in the Arctic is produced, mainly, by ice cracking which can be due both to the pressure and to thermal actions. Powerful pulses of thermal ice cracking have considerable high-frequency components, which generate a high-level noise for ultrasonic devices which operate at 20-40 kHz. The amplitude of these pulses may exceed the mean noise level by 40 dB. The time distribution of these pulses is close to Poissonian.

Accepting 0.015 Pa as the maximal amplitude of under-ice noise in the operating frequency band of the system, we have derived an acoustic level of the APS source of 166 dB (re. 1 μPa at 1 m), which should provide an SNR of 10 dB at a 1 km range. For reliable signal reception and processing in the APS components spanning approximately 800 m, we propose to increase the acoustic level of the APS sources up to 180 dB. For a

deep-water system with a 2.5 km span of the elements, this value should be, at least, 190 dB.

2. Quality of the calibration.

Calibration accuracy strongly affects the resulting accuracy of positioning. The error in horizontal positioning of the source as a function of the error in horizontal positioning of the single beacon, is shown in Fig. 9. In calculations we assumed a symmetrical APS geometry.

The accuracy of the calibration of the beacon's array depends on the correctness of the sound speed data near the bottom. For accurate calibration of the beacon's array, it is necessary to measure the sound speed near the bottom during the period of APS deployment. The accuracy of sound speed measurements should be no worse than 0.5 m/s.

It is extremely difficult and, most likely impossible, to deploy the beacons from the drifting ice on to the bottom at horizontal positions absolutely symmetrical, relative to the mooring anchor. Therefore, we propose to use an additional beacon deployed at the mooring anchor, which should significantly improve the accuracy of the array calibration.

Additional errors in the array calibration could be produced by horizontal deviations in the beacon's hydrophones due to changes in the water current. The mechanical design of the beacon-transponder implies the use of a flexible connection between the hydrophones and the electronic housing combined with the beacon's anchor. Such a mooring system with an apex float allows us to deploy the hydrophones at the required altitude of 8-10 m above the bottom. The horizontal deviation of the mooring apex due to an assumed water current of 0.2 m/s, would be up to 0.7 m.

The depth of the beacons is expected to be determined with an accuracy better than 1 m. Such resolution in the depth measurements could be provided by a pressure sensor with a 0.1% relative accuracy. The contribution of the errors in the depth measurements and in the telemetry channel will not induce an error in the horizontal positioning of more than 0.5 m.

Generally, errors less than 1.9 m in the horizontal location of the beacons will not induce an error in the horizontal positioning of the source of more than 1.3 m.

3. Instrumental accuracy.

The instrumental accuracy of the APS depends mostly on electronic errors in travel time measurements, which are due, primarily, to a discrepancy of timing in the MIR and the transponders. The measurement precision is also restricted by fluctuations of the thresholds in the comparators of the receiving channels, and uncertain delays in the electronic circuits.

In the proposed system, the maximal error of the travel time measurements due to a discrepancy in the system clocks , should be less than 30 μsec. The uncertainty of time delays in the electronic circuits should not exceed 0.6 μsec.

The integral effect of the instrumental errors on to the navigation accuracy has been experimentally tested on the analogous system deployed in Lake Baikal. Figure 8 shows the distribution of fluctuations in the measured travel times MIR-T-MIR relative to the mean value. The measurements have been made for slant distances of 600 m. The number of measurement cycles was 1000. As follows from the plot, the standard deviation of the travel time measurements is about 10 μsec, which corresponds the RMS error of 1.5 cm in horizontal positioning.

In short , in the proposed system, the total error of the source positioning due to all of the above mentioned reasons, should not exceed 2.5 m, even in the case of

extremely unfavorable combination of the effects. The most probable level of the error is expected to be ~ 1 m.

Deployment of the APS

Since the ACOUS transmitting complex is planned for installment on to the Arctic bottom from the drifting ice, we propose to use free-falling beacons, which will considerably simplify the installation procedure. The sketch of such a beacon is illustrated in Fig. 10. This construction allows to use a narrow ice hole, 25 cm in diameter, for deploying the beacon under the ice, and provides fast free-falling of the system with a velocity of ~ 2 m/sec. The whole system of beacons has a weight of about 55 kg on land, and 25 kg in the water. For the pre-deployment tests we propose to use hand-driven winches with a load capacity of about 50 kg.

Deployment of the beacons should follow the completion of the installation of the main mooring system with the acoustic source and the MIR. The beacons should be dropped down to the bottom from the appropriate points on the ice floe within a definite time lag. While falling, the beacons may shift on the horizontal plane relative to the initial position, due to the water current. If the current velocity is less than 0.2 m/s, this shift should not exceed 60 m. An appropriate initial offset in the location of the holes for the beacons, relative to the main mooring system could allow us to minimize violation of the APS symmetry.

We plan an additional preliminary test of the positioning system to arrange its components, just before the final deployment.

The preliminary test includes:

–measurements of the sound speed profile from the surface to the bottom;

–tuning of the thresholds in the receiving circuits of the MIR and the beacons;

–determination of the appropriate altitude of the beacon's hydrophones above the bottom

–complete test of the APS with the beacons suspended from the ice close to the bottom.

TECHNICAL DATA

Beacon-transponder:

Acoustic power level:	180 dB re 1μPa at 1 m;
Receiving frequency:	32-34 kHz;
Reply frequency:	28 kHz;
Life time:	24 months and 3,000 replies with a manganese battery pack 165Y(145Y);
Diameter:	0.22 m;
Weight:	55 kg (in air), 25 kg (in water).

Module of Interrogator-Receiver:

Acoustic power level :	180 dB re 1μPa at 1 m;
Receiving Frequency:	27-29 kHz;
Reply Frequency:	32.768 kHz;
Life time:	Depends on the power capacity of the batteries intended for operation of the ACOUS source (or 24 months and 3,000 replies with a manganese battery pack 165Y(145Y));

II. PRINCIPLES OF DIGITAL COMMUNICATION THROUGH A DISTANT HYDROACOUSTIC CHANNEL IN THE ACOUSTIC SYSTEM OF OCEAN THERMOMETRY.

Installation of an emitting system with a low-frequency, high-powered acoustic source is one of the most difficult problems in creating an acoustic thermometry system in the Arctic Ocean in the frameworkof the ACOUS program. Obviously, a source with a cable link to a coastal controlling station, for the power supply and communications, is extremely complicated in the conditions in the Arctic. Therefore, the use of an autonomous emitting system seems to be much more preferable in the Arctic, since it should considerably simplify the procedure of source deployment, and reduce the cost of system installation and operation. However, it is very desirable, and even necessary in some cases, to have a communication channel from the emitting source to the shore. Such a channel would allow monitoring of some of the emitting system parameters in real time, for example, the status of the system batteries, or some hydrological data at the source location. In the case of deep-water mooring of the source, it is necessary to communicate the real-time data of the acoustic positioning system which should track horizontal deviations of the source.

Acoustic thermometry of the ocean implies the establishment of a trans-ocean acoustic channel for regular signal transmissions from the emitting system to the receiving array, and measurment of the signal propagation times and amplitudes. On the other hand, that channel could be used as communication source, and those signals could transfer additional information from the emitting site. Moreover, the tomographic signals applied to acoustic thermometry, have perfect features for communication.

An acoustic digital communication channel can be designed and incorporated into the acoustic thermometry system. The phase-modulated tomographic signals encoded by the Maximal Length Sequences (M-sequences) are close to optimal for digital acoustic communications. Such signals have ideal correlation characteristics, making it possible to separate different arrivals in a multi-mode (or multi-ray) signal, and measure the modal amplitudes and propagation times. These measurements provide both the primary information for acoustic thermometry, and the data on the channel transfer characteristics which are necessary for matched processing of the communication signals.

The M-sequences have a periodic structure. Generally, the acoustic transmissions in the tomographic measurements include several periods of the M-sequence. The communication data (CD) can be inserted into the last period (or few periods) by means of additional coding. Decoding of the CD should be performed after matched processing of the multi-arrival signals. Such a communication technique cannot provide a rapid flow of information, but it allows us to maximize the reliability of data reception, and to minimize the energy required for transmission of 1 information bit at a fixed probability of error.

In this report, we outline the main theoretical principles of designing a long-range hydroacoustic communication system combined with an acoustic tomography complex. By means of statistical synthesis, we will determine the optimized algorithm of signal processing in such systems. Using the additive limit approximation and the Chernov inequality, we will derive the analytical equation for the probability of error, and analyze the noise immunity of the system, following which, we will consider particular problems in the design of the system: the choice of the type of communication signals, and the signal processing algorithm based on the Hadamard Transformation.

The model of the channel.

Generally, the real hydroacoustic channel in the ocean has a complicated transfer function. The waveguide dispersion causes a multi-arrival structure of propagated broadband signals. At low frequencies each arrival in the signal form corresponds to the acoustic mode, or the modal group. The dispersion also leads to time spreading of individual arrivals, and distortion of the form of the arrival. Generally, this spreading is not identical for different arrivals. However, if the signal frequency band is sufficiently narrow, pulse spreading becomes negligible relative to the pulse width.

At this stage of study we use a simple model of the communication channel, which implies the multi-arrival structure of the signal, but ignores the different time spreading of the arrivals. In the case of a high-dispersive channel, one could use the technique of complete matched signal processing.

In our model of the communication channel, the signal is assumed to have the following form at the receiver:

$$\begin{aligned} Y(t) &= \sum_{j=0}^{r} h_j X(t-\tau_j) + N(t), \\ X(t) &= \sum_{i=0}^{k} S^{(m)}(t - i * T_s), \end{aligned} \qquad (1)$$

where $Y(t)$ is the complex envelope of the entire signal; h_j are independent stochastic amplitudes distributed normally with mean values m_j, dispersions σ_j^2, and real and imaginary components statistically independent; $X(t-\tau_j)$ is the complex envelope of the particular signal arriving with time delay τ_j; r is the number of the signal arrivals; N (t) is the complex envelope of an additive white noise with spectral density N_0; $S^{(m)}(t)$ is the information signal (IS) carrying the message of number m; T_s is the width of one bit in the IS; k is the number of bits in the IS (the IS length).

The statistical independence of the signal components allows us to turn to the discrete form of the model (1):

$$Y_i = \sum_{j=0}^{r} h_j X_{i-j} + N_i, Y_i \equiv Y(t_i); \; N_i \equiv N(t_i); \; X_{i-j} \equiv X(t_i - \tau_j), \tag{2}$$

which simplifies the synthesis and analysis of the optimized receiving algorithm in the case under consideration. The use of the discrete model (2) allows us to avoid, in the analysis, such a complex conception as an absolute continuity of probability measures.

Statistical synthesis of the signal reception algorithm.

Selection of an optimizing criterion is the second step in the development of the optimal algorithm of signal processing, following modelling of the channel. If the transmitted symbols ("ones' and "zeros") are equally probable, the criterion of "a posteriori" probability maximum coincides with the less complicated criterion of likelihood maximum and the simple criterion of minimal error probability. Such is the case examined below.

Let us consider the likelihood operator-functional $P(Y_0^l / m)$ which is the measure of the conditional probability of the discrete stochastic process $Y_0^l \equiv \{Y_i = Y(t_i); \; t_i \in [0, t_l]; \; i = 0, 1, \ldots l\}$ under the conditions of the transmitted IS of number m:

$$P(Y_0^l / m) = \int_{-\infty}^{+\infty} \ldots \int_{-\infty}^{+\infty} P(Y_0^l / h_0^r, m) P_a(h_0^r) dh_0^r \tag{3}$$

where $P_a(h_0^r)$ is the "a posteriori" probability density of the random vector $h_0^r = \{ h_j, \; j = 0 \div r \}$. For a discrete model (2), the equation for conditional probability $P(Y_0^l / h_0^r, m)$ is as follows:

$$P(Y_0^l / h_0^r, m) = \frac{\exp\{-\frac{1}{2\sigma_w^2} \sum_{i=0}^{l} \left| Y_i - \sum_{j=0}^{r} h_j S_{i-j}^{(m)} \right|^2 \}}{(2\pi \, \sigma_w^2)^n}, \tag{4}$$

where $\sigma_w^2 = N_0/(2dt)$ is the variance of discrete samples of noise N (t), N_0 is the one-sided spectral density of the noise, dt is the sampling interval, and n is the number of samples within the interval $[0,t_1]$.

Let us assume now, that the impulse transfer characteristic of the channel has been determined in tomographic measurements. Since optimal estimates should, theoretically, have a normal, or asymptotically normal distribution, one can formulate the following equation for the "a posteriori" probability density:

$$P_a(h_0^r) = \frac{\exp\{-\frac{1}{2\sigma_j^2}\sum_{j=0}^{r}\left|h_j - \bar{h}_j\right|^2\}}{(2\pi\,\sigma_j^2)^r}, \tag{5}$$

where $\bar{h}_j$ is the estimate of the random parameter h_j; σ_j is the standard deviation from the estimate. Substituting (4) and (5) into (3), we obtain:

$$P(Y_0^l/m) = \int_{-\infty}^{+\infty}\cdots\int_{-\infty}^{+\infty}\frac{\exp\{-\frac{1}{2\sigma_w^2}\sum_{i=0}^{l}\left|Y_i - \sum_{j=0}^{r}h_j S_{i-j}^{(m)}\right|^2\}}{(2\pi\,\sigma_w^2)^n}\;\frac{\exp\{-\frac{1}{2\sigma_j^2}\sum_{j=0}^{r}\left|h_j - \bar{h}_j\right|^2\}}{(2\pi\,\sigma_j^2)^r}\,dh_0^r \tag{6}$$

Performance of r separate integrations yields:

$$P(Y_0^l/m) = \frac{\exp\{-\frac{1}{2\sigma_m^2}\sum_{i=0}^{l}\left|Y_i - \sum_{j=0}^{r}\bar{h}_j S_{i-j}^{(m)}\right|^2\}}{(2\pi\,\sigma_m^2)^n}, \tag{7}$$

where $\sigma_m^2 = \sigma_w^2 + \sum_{j=0}^{r}\left|S_{i-j}^{(m)}\right|^2\sigma_j^2 = \sigma_w^2 + 2E_s^{(m)}T_s\sum_{j=0}^{r}\sigma_j^2$; and $E_s^{(m)}$ is the energy of the IS of the number m.

In accordance with the maximal likelihood criterion, the maximal value of $P(Y_0^l/m)$ is in agreement with the condition of the signal $S^{(m)}$ being transmitted through the channel. In exploring the number m which corresponds to the likelihood maximum, it

is correct to analyze any monotonous function of $P(Y_0^l / m)$. Turning back to the continuous model, one can define the following measure, the minimum of which corresponds to the reception of the signal $S^{(m)}$:

$$-\int_0^{t_l}\left|Y(t)-\sum_{j=0}^{r}\bar{H}_j S^{(m)}(t-\tau_j)\right|^2 dt = -\int_0^{t_l}|Y(t)|^2 dt + 2\mathrm{Re}\sum_{j=0}^{r}\bar{H}_j\int_0^{t_l} Y^*(t)S^{(m)}(t-\tau_j)dt -$$

$$\sum_{j=0}^{r}\sum_{\tilde{j}=0}^{r}\bar{H}_j\bar{H}_{\tilde{j}}^*\int_0^{t_l}\int_0^{t_l-\tau} S^{(m)}(t-\tau_j)S^{(m)*}(t+\tau-\tau_{\tilde{j}})dtd\tau \qquad (8)$$

If the information signals transmitted through the channel have an ideal autocorrelation function close to a Dirac delta function (pseudonoise signals), and an identical energy level, the last term in (8) does not depend on the IS number and, thus, the detection of the transmitted IS of the number m is performed by searching for the maximum of the following integral:

$$\Lambda^{(m)}(Y_0^{t_l}) = 2\mathrm{Re}\{\sum_{j=0}^{r}\bar{H}_j\int_0^{t_l} y^*(t)S^{(m)}(t-\tau_j)dt\} \qquad (9)$$

Thus, an optimized receiver calculates the m values of $\Lambda^{(m)}(Y_0^{t_l})$ by formula (9), determines the maximal value, and makes the conclusion on the number of the informational signal transmitted through the channel.

Analysis of the noise immunity of the system

Using the Chernov inequality and the upper additive limit, one can derive a simple, but reasonably exact equation for the probability of error in the receiving system. The technique of noise immunity calculations based on the Chernov inequality [1,2], allows us to solve the particular problem considered here. As was done in the synthesis of the optimal receiving system, we will use the discrete model (2) for noise immunity analysis.

Let us assume that only two informational signals $S^{(0)}(t)$ and $S^{(1)}(t)$ may be transmitted through the multi-path channel. In this "bisignal" case, the decision should be made in accordance with the meaning of the logarithm of the likelihood ratio:

$$L(Y_0^{t_l}) = \ln\left(\frac{P(Y_0^{t_l} / m=1)}{P(Y_0^{t_l} / m=0)}\right). \tag{10}$$

If $L(Y_0^{t_l}) < 0$, we accept the signal $S^{(0)}(t)$ being received, while $L(Y_0^{t_l}) > 0$ means the reception of the signal $S^{(1)}(t)$. The key role in the Chernov inequality is rendered by the logarithm of the generating function of the likelihood ratio logarithm: $\mu(s) = E(e^{sL} /\ m=0)$, which can be calculated from the equation:

$$\mu(s) = \ln \int_{-\infty}^{+\infty} P^{s}(Y_0^{l} / m=1) P^{1-s}(Y_0^{l} / m=0) dY_0^{l} =$$

$$\ln \int_{-\infty}^{+\infty} \prod_{i=1}^{n} P^{s}(Y_i / Y_0^{i-1}, m=1) P^{1-s}(Y_i / Y_0^{i-1}, m=0) P^{s}(Y_0 / m=1) P^{1-s}(Y_0 / m=0)) dY_0^{l} = \tag{11}$$

$$\ln \int_{-\infty}^{+\infty} \prod_{i=0}^{l} \frac{\exp\left(-\frac{\left|Y_i - \sum_{j=0}^{r} \vec{H}_j S_{i-j}^{(1)}\right|^2}{2\sigma^2} s\right)}{(2\pi\sigma^2)^{s}} \frac{\exp\left(-\frac{\left|Y_i - \sum_{j=0}^{r} \vec{H}_j S_{i-j}^{(0)}\right|^2}{2\sigma^2} (1-s)\right)}{(2\pi\sigma^2)^{1-s}} dY_i$$

Regrouping of the terms in (11) yields:

$$\mu(s) = \ln \int_{-\infty}^{+\infty} \prod_{i=0}^{l} \frac{\exp\left(-\frac{\left|Y_i - \left(\sum_{j=0}^{r} \vec{H}_j S_{i-j}^{(1)} s + \sum_{j=0}^{r} \vec{H}_j S_{i-j}^{(0)} (1-s)\right)\right|^2}{2\sigma^2}\right)}{2\pi\sigma^2} dY_i$$

$$\exp\left(-\frac{\left|\sum_{j=0}^{r} \vec{H}_j S^{(1)}(t-\tau_j) - \sum_{j=0}^{r} \vec{H}_j S^{(0)}(t-\tau_j)\right|^2}{2\sigma^2} s(1-s)\right) \tag{12}$$

After integration, the equation (12) can be considerably simplified and transformed into the following equation:

$$\mu(s) = \ln\prod_{i=0}^{l}\exp\left(-\frac{\left|\sum_{j=0}^{r}\vec{H}_j S^{(1)}(t_i-\tau_j) - \sum_{j=0}^{r}\vec{H}_j S^{(0)}(t_i-\tau_j)\right|^2}{2\sigma^2} s(1-s)\right) =$$

$$\ln\exp\left(-\frac{s(1-s)}{N_0}\int_0^{t_l}\left|\sum_{j=0}^{r}\vec{H}_j S^{(1)}(t-\tau_j) - \sum_{j=0}^{r}\vec{H}_j S^{(0)}(t-\tau_j)\right|^2 dt\right) = \quad (13)$$

$$-\frac{s(1-s)}{N_0}\int_0^{t_l}\left|\sum_{j=0}^{r}\vec{H}_j S^{(1)}(t-\tau_j) - \sum_{j=0}^{r}\vec{H}_j S^{(0)}(t-\tau_j)\right|^2 dt$$

In further calculations, we will assume that the IS has a delta correlation function and a phase modulation without loss. In that case, the equation (13) is transformed into:

$$\mu(s) =$$

$$-\frac{s(1-s)}{N_0}\sum_{j=0}^{r}\sum_{\tilde{j}=0}^{r}\vec{H}_j\vec{H}_{\tilde{j}}^{*}\left(\int_0^{t_l} S^{(0)}(t-\tau_j)S^{(0)*}(t-\tau_{\tilde{j}})dt + \int_0^{t_l} S^{(1)}(t-\tau_j)S^{(1)*}(t-\tau_{\tilde{j}})dt\right) = \quad (14)$$

$$-s(1-s)\frac{2(E_s^{(0)}+E_s^{(1)})}{N_0}\sum_{j=0}^{r}\left|\vec{H}_j\right|^2 = -s(1-s)\frac{4E_\Sigma}{N_0},$$

where E_Σ is the total energy of the signal.

In accordance with theory [4], in the presence of two equally probable hypotheses(m=0 and m=1), the probability of error in the Chernov approximation can be expressed as:

$$P_2(e) \cong \frac{\exp(\mu(\tilde{s}))}{2\tilde{s}(1-\tilde{s})\sqrt{2\mu''(\tilde{s})}},$$

(15)

where $\tilde{s}$ obeys the condition $\mu'(\tilde{s}) = 0$.

The first and second derivatives of the generating function logarithm $\mu(s)$ in (15) are determined from (14):

$$\mu'(s) = (2s-1)\frac{4E_\Sigma}{N_0}, \qquad \mu''(s) = \frac{8E_\Sigma}{N_0}. \quad (16)$$

The solution of (16) yields

$$\bar{s} = 0.5; \quad \bar{s}(1-\bar{s}) = 0.25; \quad \mu'(\bar{s}) = -\frac{E_\Sigma}{N_0}; \quad \mu''(\bar{s}) = \frac{8E_\Sigma}{N_0},$$

and the final equation for the probability of error in the simplest "bisignal" transmission:

$$P_2(e) \cong \frac{\exp(-h)}{2\sqrt{\pi\ h}}, \tag{17}$$

where $h = E_\Sigma / N_0$ is the signal-to-noise ratio (SNR) after pulse compression and match summing of the signal arrivals.

If the receiver detects one of the M potential signals transmitted with a low information rate, a satisfactory estimate of the probability of error follows from the top additive limit(see Fig. 11):

$$P_M(e) \le (M-1)P_2(e) = \frac{(M-1)\exp(-h)}{2\sqrt{\pi\ h}} = \frac{(M-1)\exp(-qB_s)}{2\sqrt{\pi\ qB_s}}, \tag{18}$$

where $q = \frac{P_s}{P_n}$ is the SNR in the effective frequency band of the signal, and B_s is a signal base (the product of the signal period and the frequency band).

The probability of error is, generally, represented for one information bit, i.e:

$$P_b(e) \le \frac{(M-1)\exp(-h)}{4\sqrt{\pi\ h}} = \frac{(M-1)\exp(-qB_s)}{4\sqrt{\pi\ qB_s}}. \tag{19}$$

It is useful also to relate the probability of error with the information rate $R = \ln(M)/T_s$ and the channel throughput $C = h/T_s$, without restriction to the frequency bandwidth:

$$P_b(e) \le \frac{\exp(-T_s(C-R))}{4\sqrt{\pi\ T_s C}}. \tag{20}$$

The following equation connects the probability of error with the energy $h_b = h/\ln(M)$ required for transmitting 1 information bit:

$$P_b(e) \le \frac{\exp(-T_s R(h_b - 1))}{4\sqrt{\pi\ T_s R\ h_b}} \tag{21}$$

Let us note, that such a high noise immunity can be obtained only for signals processed with good match to the communication channel.

Construction of the signal code

Let $c_0^{n-1} = \{c_0, c_1, ..., c_{n-1}\}$ be a pseudo-random M-sequence code of $n = 2^M - 1$ binary symbols transmitted through the channel for acoustic tomography measurements. In tomographic signals, the transmitted code, generally, controls phase modulation of the carrier frequency. Let us consider, for example, a biphase modulation. In this case, the complex envelope of the signal can be written in the following form:

$$S^{(s)}(t, c_0^{n-1}) = \sqrt{2E_s T_s} \sum_{i=0}^{n} \sigma_0(t - i\tau_s) \exp(j(1-2c_i)\pi) \tag{22}$$

where $\sigma_0(t)$ is a rectangular pulse with a unit amplitude, and duration τ_s. The periodic sequence of such signals has a cyclic correlation function which is ideal for tomographic measurements.

Let $d_0^{n-1} = \{d_0, d_1, ..., d_{n-1}\}$ be the second M-sequence code of the same length $n = 2^M - 1$, which has a generating polynomial different from that of the first M-sequence. These two M-sequences do not correlate to each other. The cyclic shift of the M-sequence by m symbols generates $n = 2^M - 1$ code combinations $\{d_m, d_{m+1}, d_{m+2}, \ldots, d_{n-m-1}\} = \{d_{(i+m) \bmod n}\}$. With the trivial all-zeros code it yields 2^M different combinations. Let us assume now, that the information intended for transmission, consists of n+1 particular messages, and number that the m $(0 \le m < n)$ relates to the number of the message.

We propose to insert the *information* code $\{d_{(i+m) \bmod n}\}$ into the last period (or the last few periods) of the *measuring* code c_0^{n-1} via logical exclusive "or" summing. The resulting code combination is:

$$e_0^n = \{e_0, e_1, e_2, \ldots, e_n\} = \{c_0 \oplus d_m, c_1 \oplus d_{m+1}, \ldots, c_n \oplus d_{n-m-1}\} = \{c_i \oplus d_{(i+m) \bmod n}\}, \quad (23)$$

where $c_i \oplus d_i$ means an exclusive "or" summation. The code combinations composed in such a way, belong to one of the classes adjacent to the Gold code [3]. None of these combinations can be formed from another one through a cyclic shift. The combined informational signal has the following complex envelope:

$$S^{(m)}(t,\ e_0^{n-1}) = \sqrt{2E_sT_s}\sum_{i=0}^{n}\sigma_o(t - i\tau_s)\exp(\,j(1 - 2(c_i \oplus d_{(i+m) \bmod n}))\pi)\,. \quad (24)$$

It is important to notice that, in multi-path channels, the measuring M-sequence code cannot be used directly for coding information in such a way, because time spreading of the signal leads to an uncertainty in decoding of the information.

The resulting sequence combined for transmission, consists of k measuring and one information, signals:

$$X(t) = \sum_{i=0}^{k-1} S^{(s)}(t - iT_s,\ \{c_0, c_1, \ldots, c_{n-1}\}\ ) + S^{(m)}(t - kT_s,\ \{c_0 \oplus d_m, c_1 \oplus d_{m+1}, \ldots, c_n \oplus d_{n-m-1}\}\ )\ . \quad (25)$$

Thus, using the proposed technique, one can transmit 2^M informational signals: $2^M - 1$ combinations generated from two M-sequences, and one "zero" code which is simply a repetition of the measuring M-sequence code.

The algorithm of signal processing

Leaving out most of the details of the commonly accepted methods for the processing of pseudo-noise signals, let us consider the key procedure of the algorithm proposed for processing of the tomographic signals, as well as for decoding of the transmitted information. Correlation decoding, commonly referred to as a pulse compression technique, is the main component of signal processing which requires most of the computational time at both steps of tomographic measurements and information decoding. In the discrete form, the procedure implies multiplication of the matrix

$S \equiv \{S_i^{(m)}\}$ made up of M different replicas of the transmitted signals, by the vector of the received signal $Y \equiv \{Y_i\}$:

$$\mathbf{Z} = \mathbf{SY} = \begin{pmatrix} Z_0 \\ Z_1 \\ \cdots \\ Z_m \\ \cdots \\ Z_{M-1} \end{pmatrix} = \begin{pmatrix} S_0^{(0)} & S_1^{(0)} & S_2^{(0)} & \cdots & S_n^{(0)} \\ S_0^{(1)} & S_1^{(1)} & S_2^{(1)} & \cdots & S_n^{(1)} \\ \cdots & \cdots & \cdots & \cdots & \cdots \\ S_0^{(m)} & S_1^{(m)} & S_2^{(m)} & \cdots & S_n^{(m)} \\ \cdots & \cdots & \cdots & \cdots & \cdots \\ S_0^{(M-1)} & S_1^{(M-1)} & S_2^{(M-1)} & \cdots & S_n^{(M-1)} \end{pmatrix} \begin{pmatrix} Y_0 \\ Y_1 \\ \cdots \\ Y_i \\ \cdots \\ Y_{n-1} \end{pmatrix}, \qquad (26)$$

where $Y_i = Y(t_i)$; $S_i^{(m)} = S^{(m)}(t_i)$; $t_i = t_0 + i\Delta t$; and Δt is the sampling interval.

At the stage of processing of the tomographic signals, the matrix S consists of n vectors-rows, each of which represents the measuring M-sequence code shifted cyclically by the particular number m of the code symbols:

$$\mathbf{Z} = \mathbf{SY} = \begin{pmatrix} Z_0 \\ Z_1 \\ \cdots \\ Z_m \\ \cdots \\ Z_{n-1} \end{pmatrix} = \begin{pmatrix} S_0 & S_1 & S_2 & \cdots & S_{n-1} \\ S_1 & S_2 & S_3 & \cdots & S_0 \\ \cdots & \cdots & \cdots & \cdots & \cdots \\ S_m & S_{m+1} & S_{m+2} & \cdots & S_{m-1} \\ \cdots & \cdots & \cdots & \cdots & \cdots \\ S_{n-1} & S_0 & S_1 & \cdots & S_{n-2} \end{pmatrix} \begin{pmatrix} Y_0 \\ Y_1 \\ \cdots \\ Y_m \\ \cdots \\ Y_{n-1} \end{pmatrix} \qquad (27)$$

Each of the n possible discrete shifts, rated by the sampling interval, corresponds to the time delay of the signal arrival. Thus, the resulting vector Z represents the impulse response of the channel in the discrete time domain, which provides the estimates of the signal arrival times τ_i and the complex arrival amplitudes h_i $(1 \le i \le r)$.

At the second stage of decoding the transmitted data, the informational signal received at the end of the whole transmitted signal, should be delayed by each of the estimated times τ_i, and multiplied, element-by-element, by the replica of the measuring signal. After that, multiplying the resulting signals by the cyclic matrix of the information

M-sequence, we obtain the correlation vectors Z_i which characterizes the correlation of the information signal delayed by τ_i with the set of orthogonal signals, which represent all possible messages scheduled for transmission.

The next step of the matched signal processing implies a coherent summation of all significant signal arrivals, i.e. the summation of correlation vectors Z_i multiplied by complex conjugate coefficients h_i^*. This procedure should optimize the SNR, and minimize the probability of error. Thus, processing of the information signals includes r matrix-by-vector multiplications which require most of the computation time.

However, the multiplication of an arbitrary vector by the cyclic matrix of the M-sequence consisting of symbols + 1 and -1, can be performed quickly via transposition of the vector elements, the Hadamard transformation (the fast Walsh transformation), and the final transposition of the resulting vector. With the use of the Hadamard transformation, one can compute both the channel transfer characteristic and the information correlation vector. Application of the Hadamard transformation will allow us to perform the signal processing algorithm without matrix operations of multiplication and division, and with the minimal number of matrix summations and subtractions, which considerably increases the efficiency of the processing algorithm.

The analogue communication technique has been successfully used for remote control and positioning of autonomous, acoustic bottom stations operated in the high-frequency band at 8.5 kHz, in the Mediterranean Sea [4].

CONCLUSION

The acoustic positioning system, proposed in the report, has been adapted to the conditions and requirements of the ACOUS experiment. The system should provide accuracy of horizontal positioning of the source not worse than 5 m, which corresponds to the temperature resolution of the thermometry measurements of about 1.5 millidegree at a 1 Mm path. The acoustic beacons are proposed to be free-falling and self-locating, allowing reliable and accurate operation of the positioning system, while, at the same time, reducing the time and cost of deployment of the system.

The communication method proposed for transmitting the navigation data, is well integrated with the acoustic thermometry technique. The method should provide reliable communication of short data series of 256 - 512 ~~Kb~~ bits, without significant increase in the power consumption of the system ordered for the tomographic transmissions.

The combination of the acoustic positioning system with the communication method considered in this report, could give us an instrument which would help us to solve one of the main problems associated with the deployment of an autonomous emitting system in the Central Arctic Basin.

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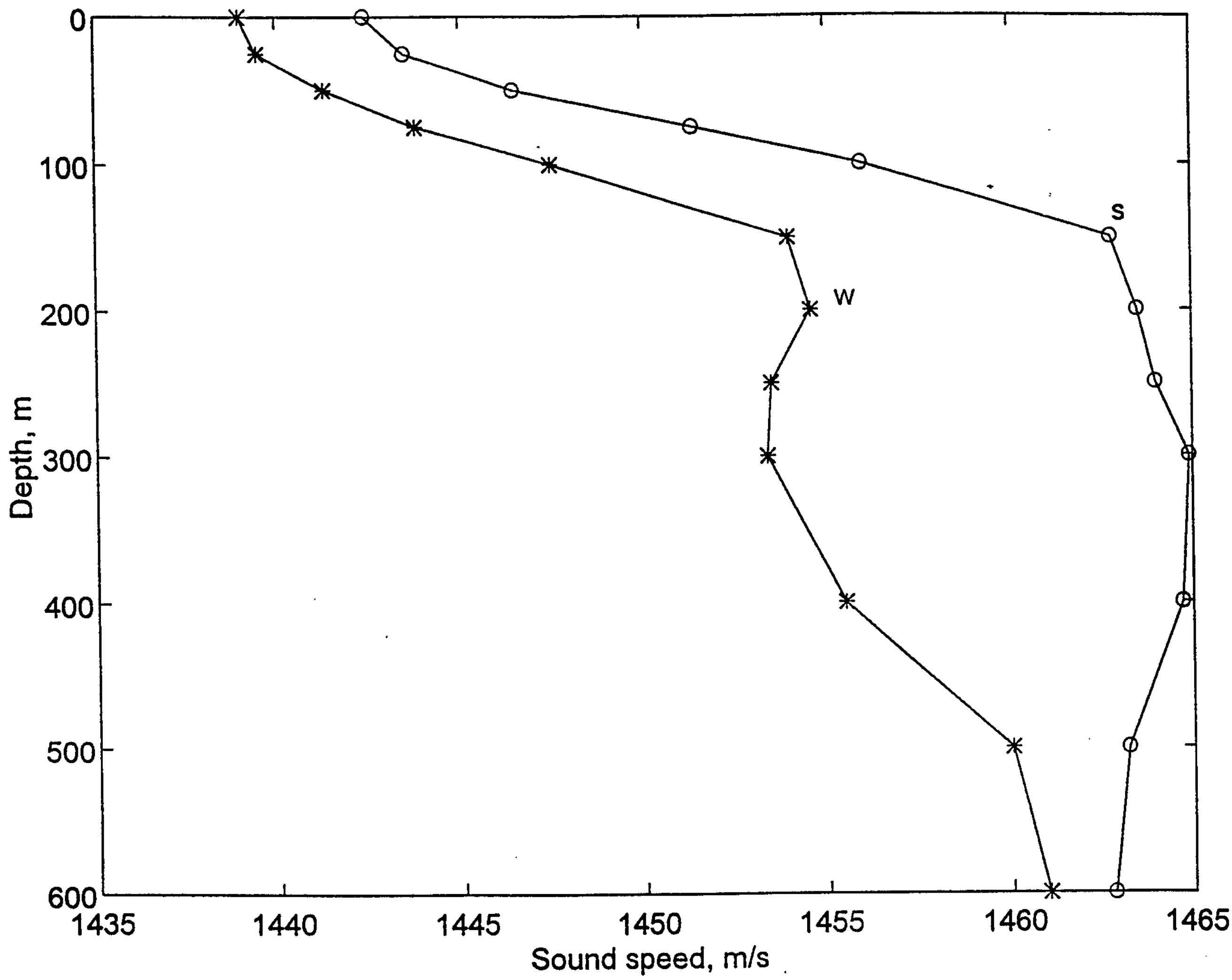

Fig. 1. Vertical profiles of the minimal (winter) and maximal (summer) sound speed values measured in the northern part of the St. Anna Strait.

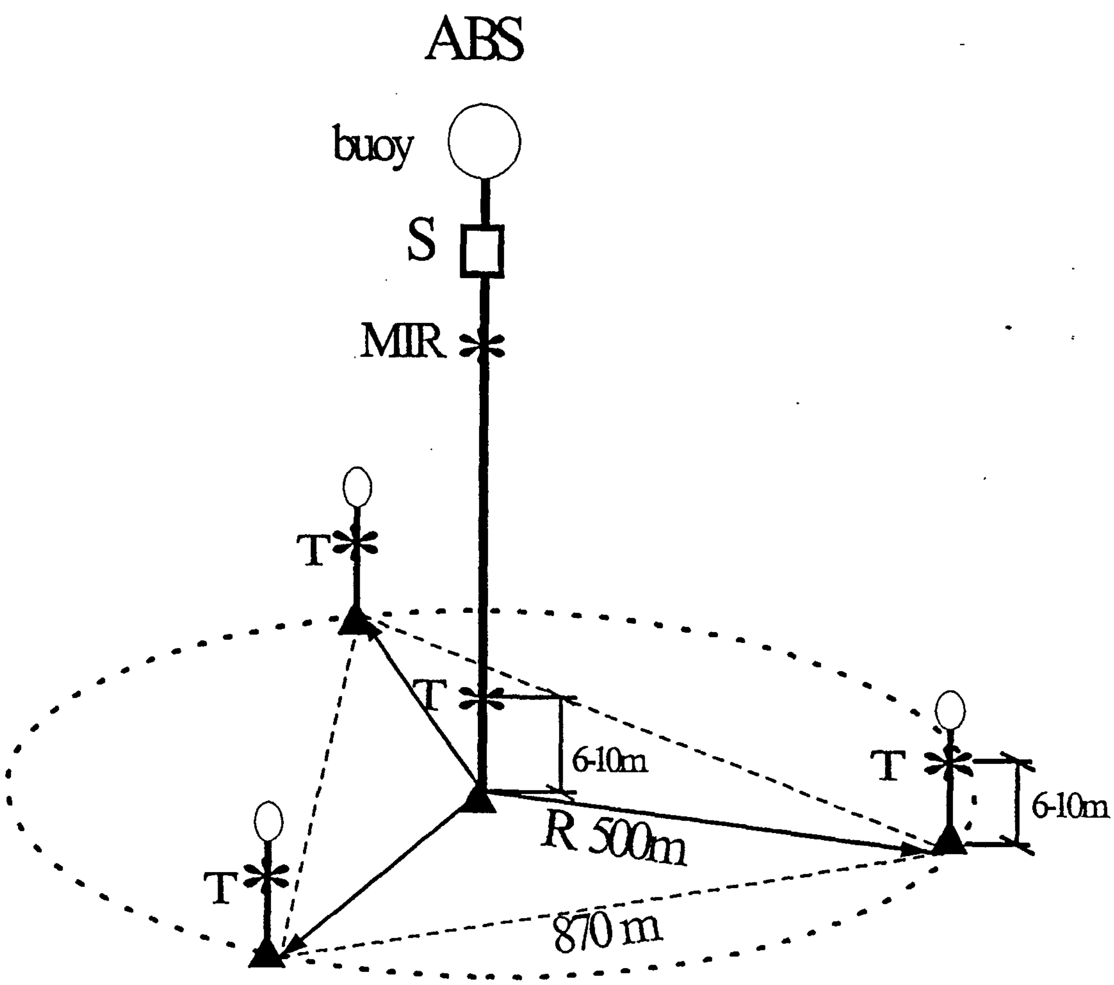

Fig.2. General scheme of the APS: S - Acoustic source; MIR - Module of the interrogator-receiver; T - Beacon-transponder

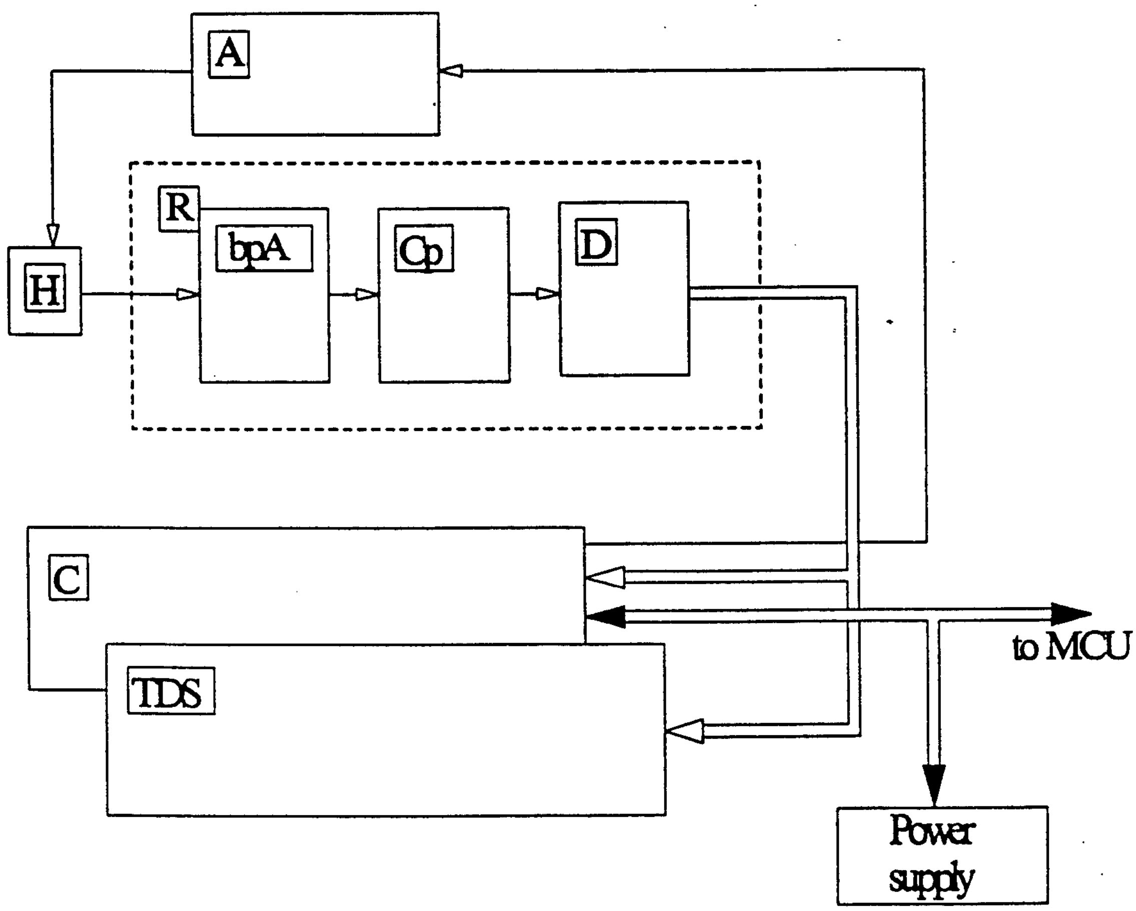

Fig.3. Functional scheme of the MIR: C - controller; TDC - time-to-digit converter; R - receiver; A - output amplifier; H - hydrophone; bpA band-pass amplifier; Cp - comparator; D - pulse duration discriminator.

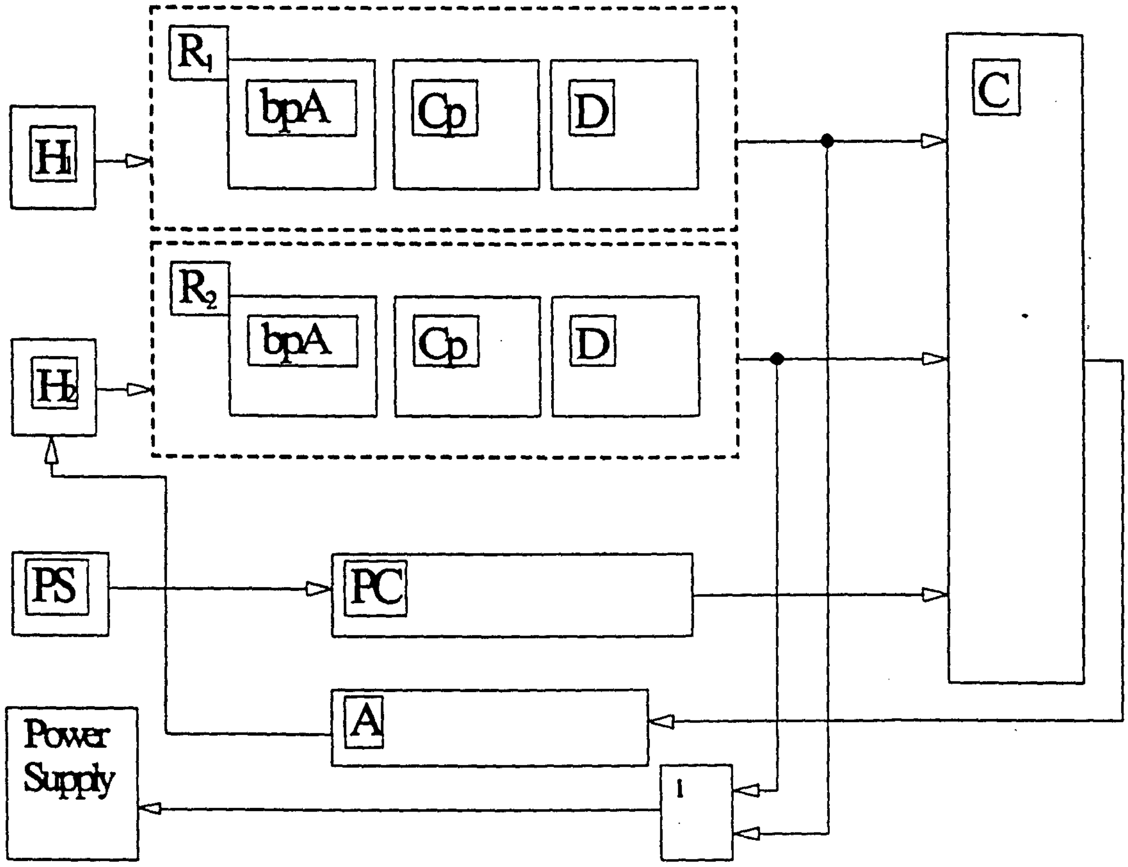

Fig.4. Functional scheme of the beacon-transponder: R1 and R2 - two receiving channels; H1 and H2 - hydrophones; PC - pressure converter; C - controller; bpA - band-pass amplifier; Cp - comparator; D - pulse duration discriminator.

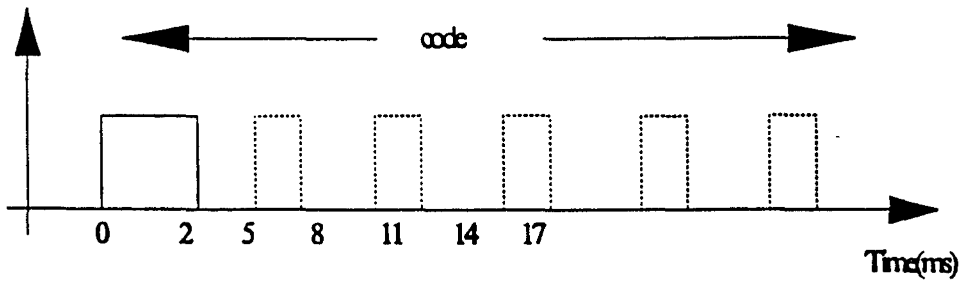

Fig.5. Time sequence of the transmitted code.

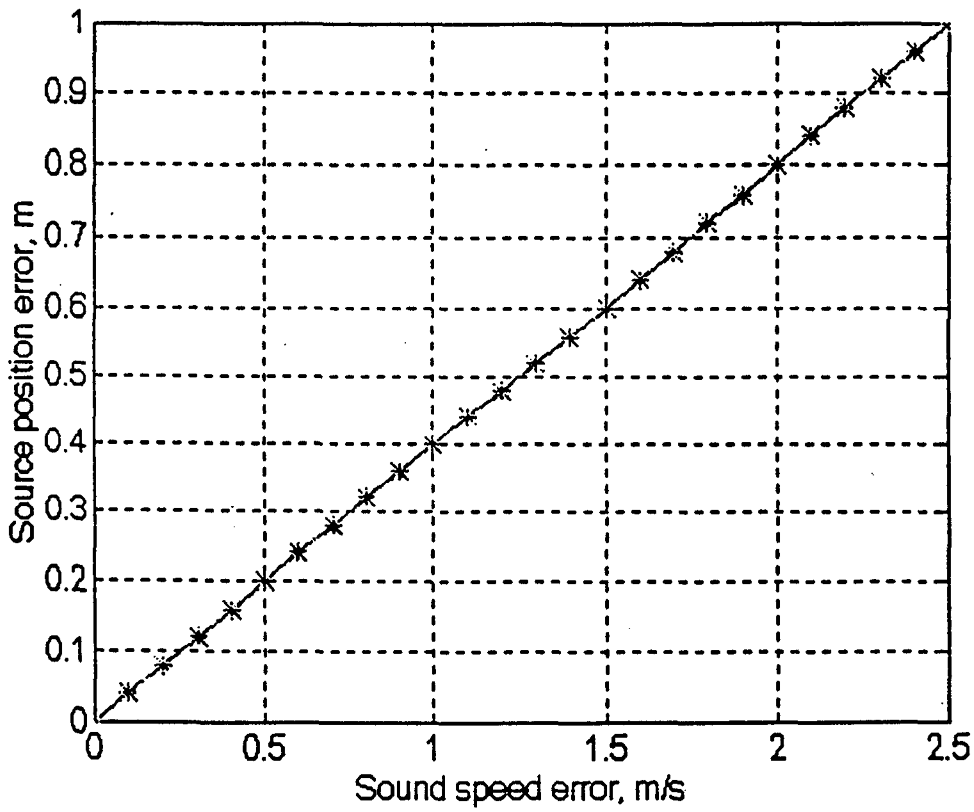

Fig.6. The error in the horizontal positioning of the source due to the error of the mean sound speed along one of three slant distances in the symmetrical system.

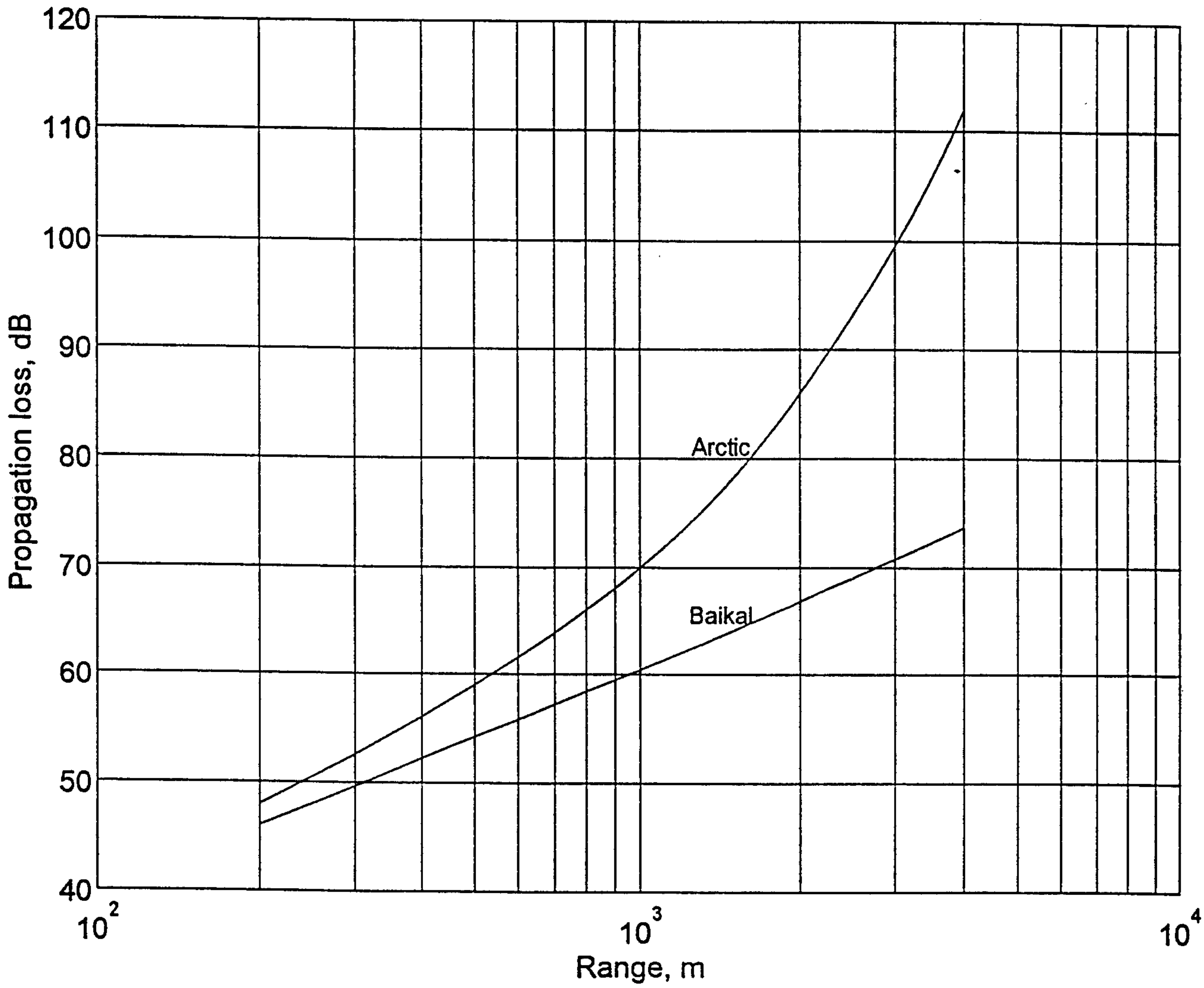

Fig.7. Attenuation of acoustic signals at 30 kHz in (1) Lake Baikal and (2) in the Arctic Ocean.

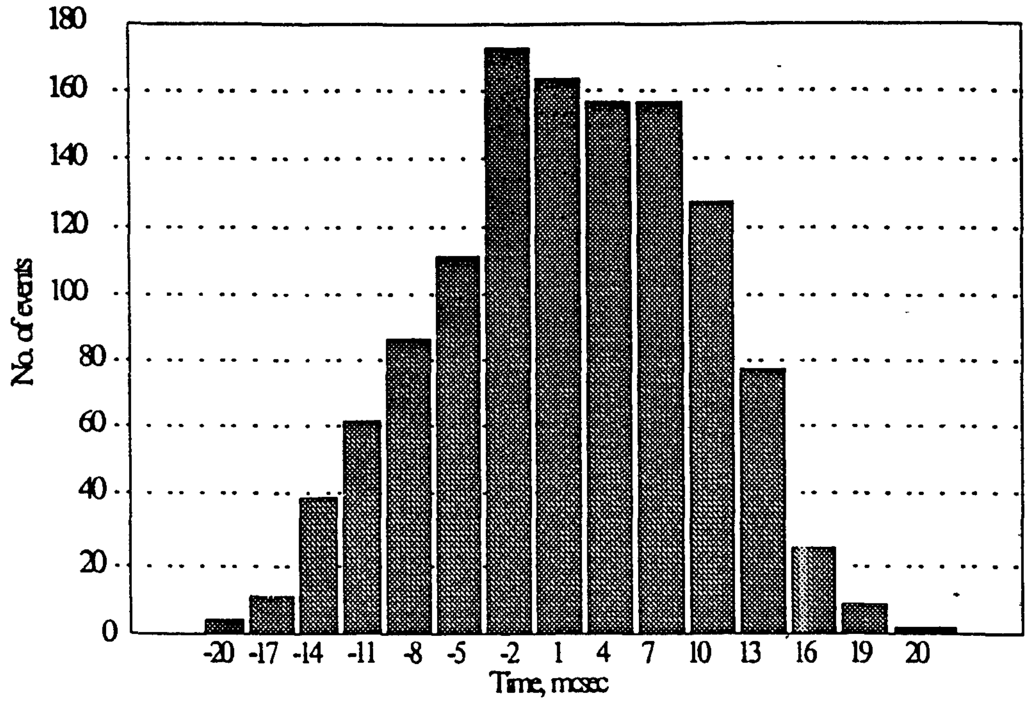

Fig.8. Distribution of the error in 1000 travel time measurements, at a range of 600 m.

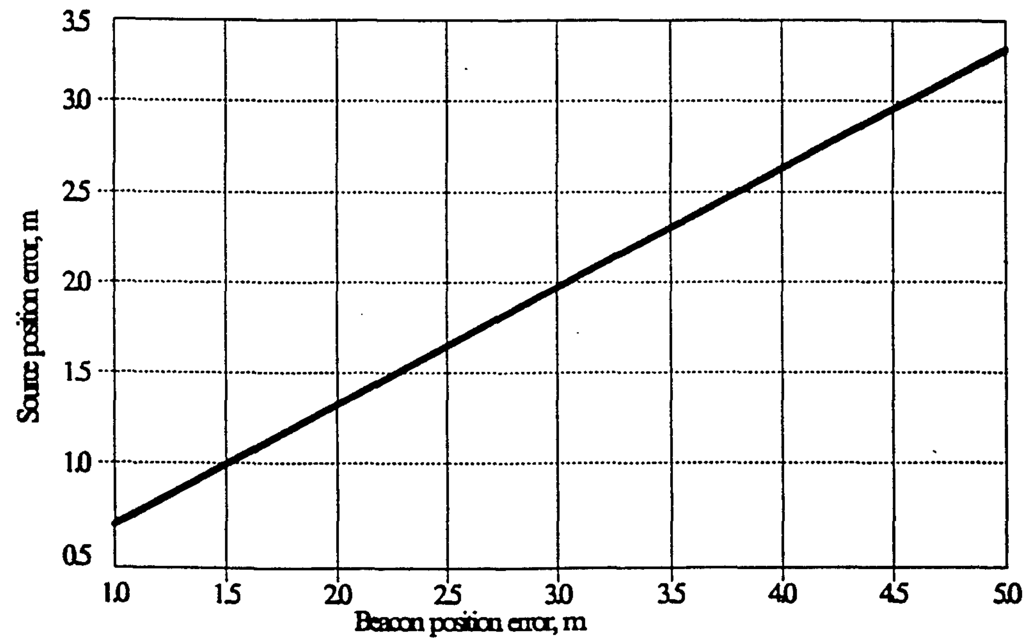

Fig.9. The error in horizontal positioning of the source due to the error of the beacon location in the symmetrical system.

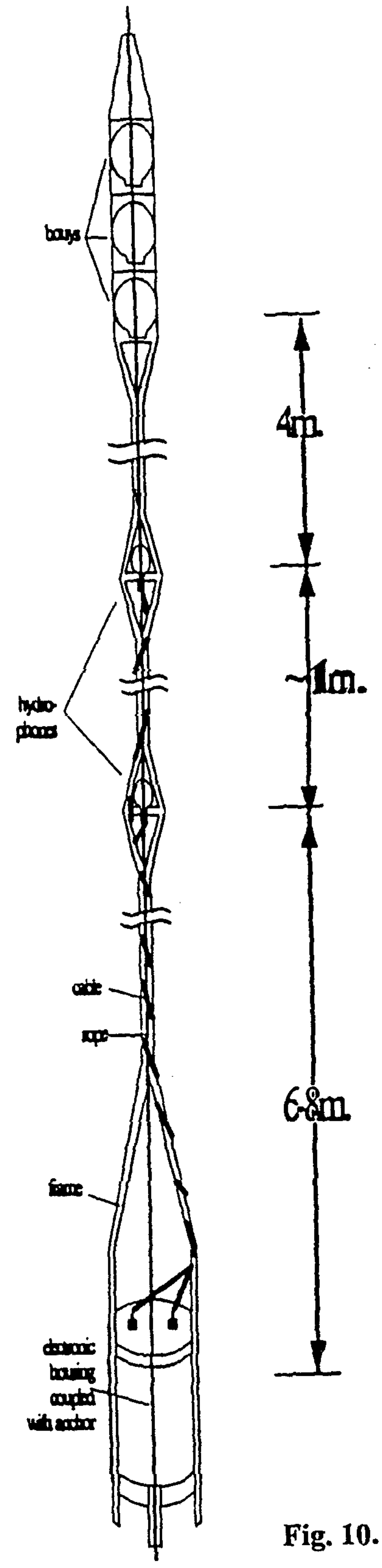

Fig. 10. Sketch of the free-falling beacon-transponder.

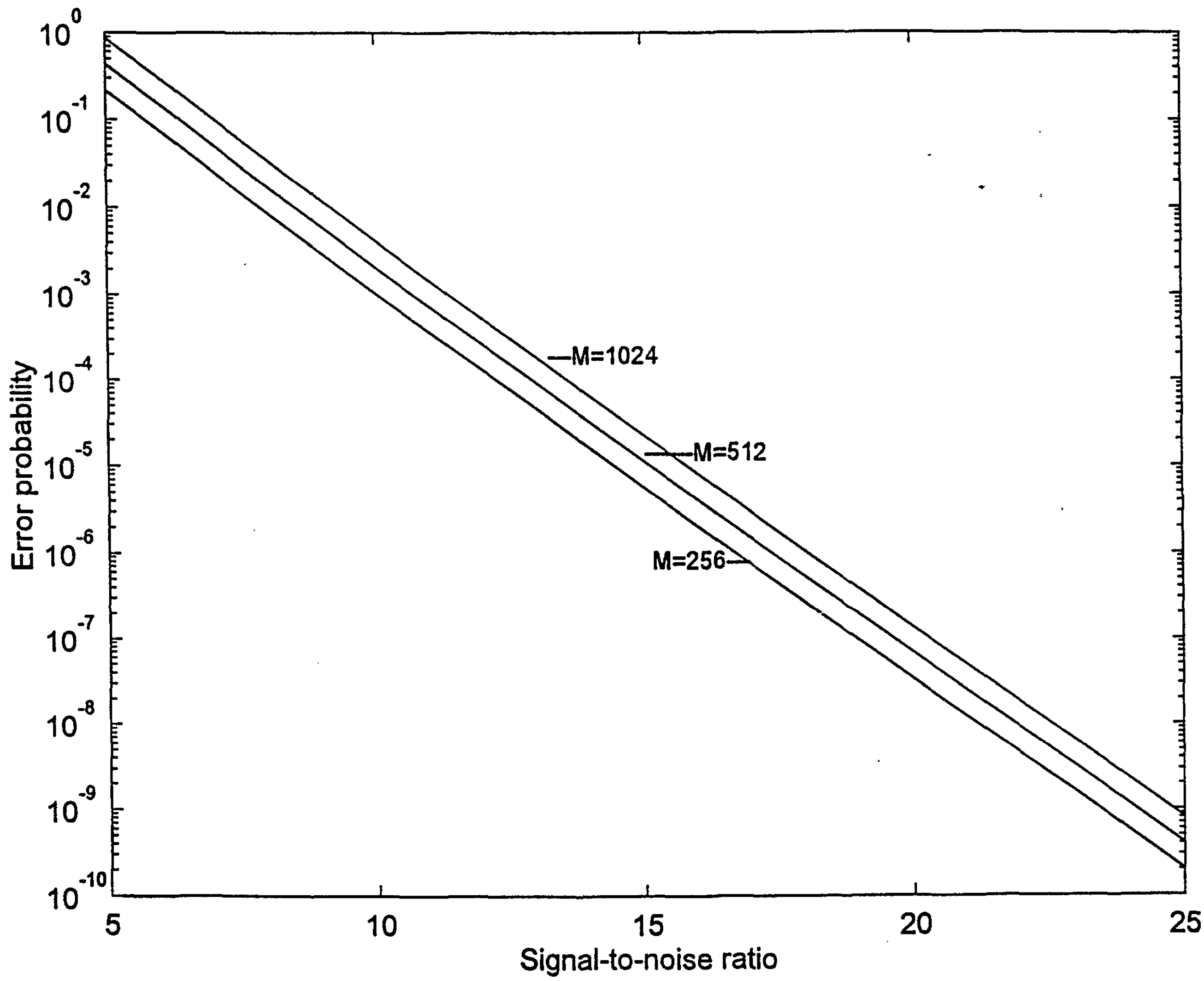

Fig. 11. Probability of the error (Eq.18) in the decoding of information signals with different numbers of the transmitted code combinations: 256, 512, and 1024, versus the signal-to-noise ratio.